podcast3

May 9, 2021

```
In [177]: import os
          from scipy.io import wavfile
          import pandas as pd
          import matplotlib.pyplot as plt
          import numpy as np
          from keras.layers import Conv2D, MaxPool2D, Flatten
          from keras.layers import Dropout, Dense, TimeDistributed
          from keras.models import Sequential
          from keras.utils import to_categorical
          from sklearn.utils.class_weight import compute_class_weight
          from tqdm import tqdm
          from python_speech_features import mfcc
          from sklearn.metrics import accuracy_score
          from keras.callbacks import ModelCheckpoint

          path = 'C://Users//richard//OneDrive//Documents//Audacity//wavfiles//'
          audio_classes_path = 'C://Users//richard//OneDrive//Documents//Audacity//audio_classe
```

0.1 Method

- Using the csv file which contains the list of files and their labels, generate a pandas dataframe.
- Use the dataframe to index the files and get a random window of audio
- Train on this audio using a CNN. Train, test split is not required as all the audio comes from the same place and is of small size
- Go through all the files every 0.1 s and compare prediction to label
- See Final preedcitons and conclusions at bottom of file.

0.2 Class distribution

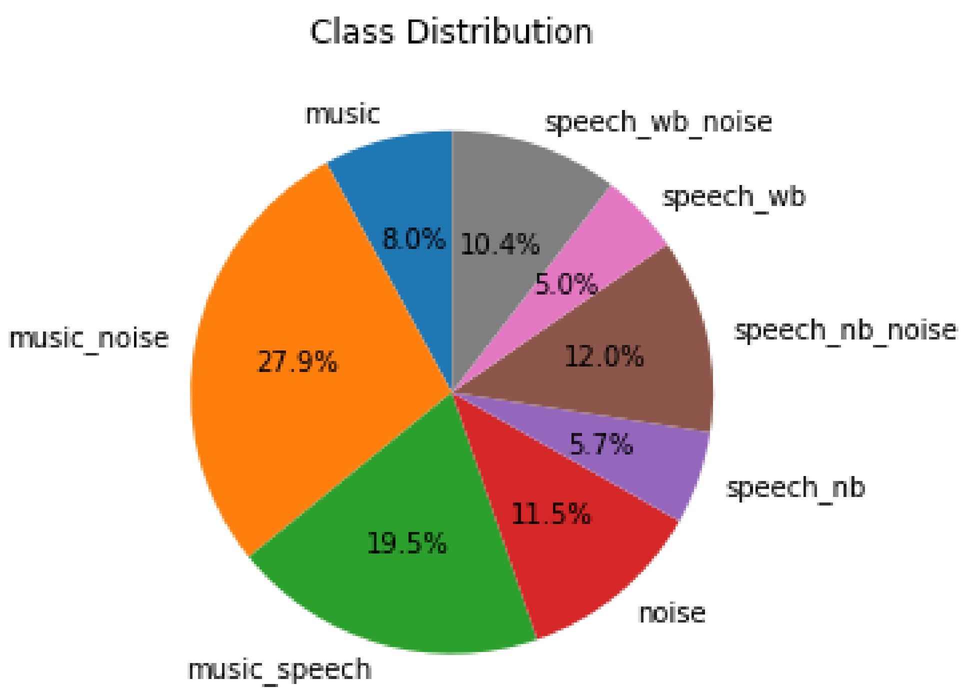
```
In [178]: df = pd.read_csv(audio_classes_path)
          df.set_index('file_name', inplace=True)

          for f in df.index:
              rate, signal = wavfile.read(path+f)
              df.at[f, 'length'] = signal.shape[0]/rate
```

```
classes = list(np.unique(df.label))
class_dist = df.groupby(['label'])['length'].mean()

fig, ax = plt.subplots()
ax.set_title('Class Distribution', y=1.08)
ax.pie(class_dist, labels=class_dist.index, autopct='%1.1f%%',
       shadow=False, startangle=90)
ax.axis('equal')
plt.show()
```

```
C:\Users\richard\Anaconda3\envs\tf15\lib\site-packages\scipy\io\wavfile.py:273: WavFileWarning
  WavFileWarning)
```



In [179]:
```
df.head()
```

Out[179]:

file_name	label	length
music_plus_noise1.wav	music_noise	7.543288
music_plus_noise2.wav	music_noise	6.991315
music_twintones1.wav	music_noise	5.944308
music_plus_speech1.wav	music_speech	4.768209
music1.wav	music	2.018050

In [180]:
```
#get number of samples to draw from the audio files
n_samples = 2 * int(df['length'].sum()/0.1)
n_samples
```

Out[180]: 1896

In [181]:
```
#total length of all the audio in the files
df.length.sum()
```

Out[181]: 94.87106575963719

In [182]:
```
#0 -1 class dist
prob_dist = class_dist / class_dist.sum()

#get a choice of instrument randomly based on class distribution
choices = np.random.choice(class_dist.index,p=prob_dist)

class_dist=class_dist.sort_values(ascending=False)
class_dist
```

Out[182]:
```
label
music_noise        6.826304
music_speech       4.768209
speech_nb_noise    2.947545
noise              2.823212
speech_wb_noise    2.554422
music              1.955057
speech_nb          1.409063
speech_wb          1.223050
Name: length, dtype: float64
```

In [183]:
```
class_dist.index
```

Out[183]:
```
Index(['music_noise', 'music_speech', 'speech_nb_noise', 'noise',
       'speech_wb_noise', 'music', 'speech_nb', 'speech_wb'],
      dtype='object', name='label')
```

0.3 Make a CNN model

Use general image processing network, increasing size of feature maps at each layer. As output is eight classes use softmax at output to get majority class probability. Max pooling can be used but images are small anyway 13x13 or 26x26. Since using small size and same data files will overfit anyway so not using test and validation set.

In [184]:
```
def get_conv_model():
    model = Sequential()
    model.add(Conv2D(32,(3,3),activation='relu',strides=(1,1),padding='same',input_sh
    #model.add(MaxPool2D((2,2)))
    model.add(Conv2D(64,(3,3),activation='relu',strides=(1,1),padding='same'))
    #model.add(MaxPool2D((2,2)))
    model.add(Conv2D(128,(3,3),activation='relu',strides=(1,1),padding='same'))
    #model.add(MaxPool2D((2,2)))
    model.add(Conv2D(256,(3,3),activation='relu',strides=(1,1),padding='same'))
```

```
        #model.add(MaxPool2D((2,2)))
        model.add(Dropout(0.1))
        model.add(Flatten())
        model.add(Dense(256,activation='relu'))
        model.add(Dense(128,activation='relu'))
        model.add(Dense(64,activation='relu'))
        model.add(Dense(8,activation='softmax'))
        model.summary()
        model.compile(loss='categorical_crossentropy',optimizer='adam',metrics=['acc'])
        return model
```

0.4 Make a helper class

```
In [185]: class Config():
              def __init__(self,mode = 'conv', nfilt=26, nfeat=26, nfft=2048,rate=16000):
                  self.mode = mode
                  self.nfilt = nfilt
                  self.nfeat = nfeat
                  self.nfft = nfft
                  self.rate = rate
                  self.step = int(rate/10)
```

0.5 Generate data from files

Need to get random chunks of the data from the files. These chunks here are 0.1s long and selected randomly from the distribution given above

```
In [186]: #generate random sampling from dataset
          def build_rand_feat():
              X = [];y=[]

              _min,_max = float('inf'),-float('inf')

              for _ in tqdm(range(n_samples)):
                  rand_class = np.random.choice(class_dist.index,p=prob_dist)

                  #get random file based on random instrument
                  file = np.random.choice(df[df.label==rand_class].index)
                  rate,wav = wavfile.read(path + file)
                  label = df.at[file,'label']

                  #get random chunk from file
                  rand_index = np.random.randint(0,wav.shape[0] - config.step)
                  sample = wav[rand_index:rand_index+config.step]
                  X_sample = mfcc(sample,rate,numcep=config.nfeat, nfilt = config.nfilt, nfft=c

                  #running update of min/max
                  _min = min(np.amin(X_sample),_min)
```

```
                _max = max(np.amax(X_sample),_max)
                X.append(X_sample)
                y.append(classes.index(label))                              #get index number

            config.min = _min
            config.max = _max

            X, y = np.array(X),np.array(y)                                  #normalise the dat

            X = (X - _min)/ (_max - _min)

            X = X.reshape(X.shape[0],X.shape[1], X.shape[2],1) #samples * rows (time) * cols
            y = to_categorical(y,num_classes=8)

            _min = str(_min)
            _max = str(_max)

            return X,y,_min,_max
```

In [187]:
```
config= Config()
```

In [188]:
```
#build dataset using distirbution we have defined
X,y,_min,_max = build_rand_feat()
y_flat = np.argmax(y,axis=1)      #get the max value across each row
input_shape = (X.shape[1],X.shape[2],1)
model = get_conv_model()
```

```
  3%|                                                                              | 63/1896 [00
  WavFileWarning)
100%|| 1896/1896 [00:05<00:00, 317.53it/s]


_________________________________________________________________
Layer (type)                 Output Shape              Param #
=================================================================
conv2d_29 (Conv2D)           (None, 3, 26, 32)         320
_________________________________________________________________
conv2d_30 (Conv2D)           (None, 3, 26, 64)         18496
_________________________________________________________________
conv2d_31 (Conv2D)           (None, 3, 26, 128)        73856
_________________________________________________________________
conv2d_32 (Conv2D)           (None, 3, 26, 256)        295168
_________________________________________________________________
dropout_8 (Dropout)          (None, 3, 26, 256)        0
_________________________________________________________________
flatten_8 (Flatten)          (None, 19968)             0
_________________________________________________________________
dense_29 (Dense)             (None, 256)               5112064
_________________________________________________________________
```

```
dense_30 (Dense)             (None, 128)              32896
_________________________________________________________________
dense_31 (Dense)             (None, 64)               8256
_________________________________________________________________
dense_32 (Dense)             (None, 8)                520
=================================================================
Total params: 5,541,576
Trainable params: 5,541,576
Non-trainable params: 0
_________________________________________________________________
```

In [189]:
```
print(X.shape)
print(y.shape)
```

```
(1896, 3, 26, 1)
(1896, 8)
```

In [190]:
```
#base weights on prob dist defined above - need as unbalanced
class_weight = compute_class_weight('balanced',np.unique(y_flat),y_flat)
```

In [191]:
```
class_weight
```

Out[191]:
```
array([1.05803571, 1.52903226, 0.43807763, 1.15048544, 2.69318182,
       0.62204724, 1.16748768, 2.41836735])
```

In [192]:
```
#try adding weights to classes or not
#model.fit(X,y,epochs=30,batch_size=32,shuffle=True,class_weight=class_weight,verbos
model.fit(X,y,epochs=50,batch_size=16,shuffle=True,verbose=2)
```

```
Epoch 1/50
 - 2s - loss: 1.8995 - acc: 0.2911
Epoch 2/50
 - 2s - loss: 0.7511 - acc: 0.7431
Epoch 3/50
 - 2s - loss: 0.5452 - acc: 0.8043
Epoch 4/50
 - 2s - loss: 0.4738 - acc: 0.8233
Epoch 5/50
 - 2s - loss: 0.3807 - acc: 0.8523
Epoch 6/50
 - 2s - loss: 0.3445 - acc: 0.8745
Epoch 7/50
 - 2s - loss: 0.3194 - acc: 0.8739
Epoch 8/50
 - 2s - loss: 0.3054 - acc: 0.8797
Epoch 9/50
 - 2s - loss: 0.2588 - acc: 0.9024
```

```
Epoch 10/50
 - 2s - loss: 0.2461 - acc: 0.9103
Epoch 11/50
 - 2s - loss: 0.2229 - acc: 0.9182
Epoch 12/50
 - 2s - loss: 0.2072 - acc: 0.9272
Epoch 13/50
 - 2s - loss: 0.2244 - acc: 0.9204
Epoch 14/50
 - 2s - loss: 0.1882 - acc: 0.9330
Epoch 15/50
 - 2s - loss: 0.1687 - acc: 0.9346
Epoch 16/50
 - 2s - loss: 0.1273 - acc: 0.9525
Epoch 17/50
 - 2s - loss: 0.1390 - acc: 0.9499
Epoch 18/50
 - 2s - loss: 0.1422 - acc: 0.9509
Epoch 19/50
 - 2s - loss: 0.1192 - acc: 0.9578
Epoch 20/50
 - 2s - loss: 0.1400 - acc: 0.9515
Epoch 21/50
 - 2s - loss: 0.1037 - acc: 0.9641
Epoch 22/50
 - 2s - loss: 0.0976 - acc: 0.9673
Epoch 23/50
 - 2s - loss: 0.1026 - acc: 0.9673
Epoch 24/50
 - 2s - loss: 0.1040 - acc: 0.9631
Epoch 25/50
 - 2s - loss: 0.1080 - acc: 0.9620
Epoch 26/50
 - 2s - loss: 0.0553 - acc: 0.9815
Epoch 27/50
 - 2s - loss: 0.0515 - acc: 0.9826
Epoch 28/50
 - 2s - loss: 0.0357 - acc: 0.9852
Epoch 29/50
 - 2s - loss: 0.0706 - acc: 0.9784
Epoch 30/50
 - 2s - loss: 0.1031 - acc: 0.9626
Epoch 31/50
 - 2s - loss: 0.0647 - acc: 0.9768
Epoch 32/50
 - 2s - loss: 0.0695 - acc: 0.9747
Epoch 33/50
 - 2s - loss: 0.0874 - acc: 0.9678
```

```
Epoch 34/50
 - 2s - loss: 0.0649 - acc: 0.9800
Epoch 35/50
 - 2s - loss: 0.0320 - acc: 0.9884
Epoch 36/50
 - 2s - loss: 0.0560 - acc: 0.9836
Epoch 37/50
 - 2s - loss: 0.0389 - acc: 0.9868
Epoch 38/50
 - 2s - loss: 0.0711 - acc: 0.9773
Epoch 39/50
 - 2s - loss: 0.0699 - acc: 0.9757
Epoch 40/50
 - 2s - loss: 0.0243 - acc: 0.9916
Epoch 41/50
 - 2s - loss: 0.0164 - acc: 0.9931
Epoch 42/50
 - 2s - loss: 0.0265 - acc: 0.9910
Epoch 43/50
 - 2s - loss: 0.0531 - acc: 0.9805
Epoch 44/50
 - 2s - loss: 0.0496 - acc: 0.9826
Epoch 45/50
 - 2s - loss: 0.1470 - acc: 0.9573
Epoch 46/50
 - 2s - loss: 0.0508 - acc: 0.9831
Epoch 47/50
 - 2s - loss: 0.0186 - acc: 0.9942
Epoch 48/50
 - 2s - loss: 0.0702 - acc: 0.9763
Epoch 49/50
 - 2s - loss: 0.0407 - acc: 0.9873
Epoch 50/50
 - 2s - loss: 0.0385 - acc: 0.9900
```

```
Out[192]: <keras.callbacks.History at 0x18d95d8a908>
```

1 Predictions 1

Use the model to get predictions, first across all files in directory

```
In [193]: def build_pred_all_files(fn2class,audio_dir=path):
              y_true = []
              y_pred = []
              fn_prob = {}

              for fn in tqdm(os.listdir(audio_dir)):
```

```
            rate, wav = wavfile.read(os.path.join(audio_dir,fn))
            label = fn2class[fn]
            c = classes.index(label)
            y_prob = []
            y_per_file=[]

            #print(wav.shape[0],config.step,fn)
            #break

            for i in range(0,wav.shape[0]-config.step,config.step):
                sample = wav[i:i+config.step]
                #sample is a block of samples x secs (0.1) long
                x = mfcc(sample,rate, numcep=config.nfeat,nfilt = config.nfilt,nfft=confi
                x = (x-float(_min))/(float(_max)-float(_min))
                x = x.reshape(1,x.shape[0],x.shape[1],1)

                #for every block of samples get the mfcc and compare to target
                #y_hat is probability for each class - softmax in final layer
                y_hat = model.predict(x)
                y_prob.append(y_hat)
                #y_pred is class corrsponding to max prob. for each block of samples (ev
                y_pred.append(np.argmax(y_hat))
                #y_true is the real class
                y_true.append(c)

            #print(label,c)
            fn_prob[fn] = np.mean(y_prob,axis=0).flatten()

        return y_true, y_pred, fn_prob
```

In [194]:
```
#y_pred is the result for each block of samples for every file
#y_true is the actual value for each block of samples for every file
#fn_prob is the value for each class probabilty for each file, the max value indicat

df = pd.read_csv(audio_classes_path)

classes = list(np.unique(df.label))

fn2class = dict(zip(df.file_name,df.label))

y_true, y_pred, fn_prob = build_pred_all_files(fn2class,path)
result = accuracy_score(y_true,y_pred)
```

```
 22%|                                                                   | 8/37 [00:03<00:13,  2.
  WavFileWarning)
 84%|             | 31/37 [00:09<00:01,  3.11it/s]C:\Users\richard\Anaconda3\envs\tf15\lib\site
  TqdmSynchronisationWarning)
100%|| 37/37 [00:11<00:00,  3.12it/s]
```

In [195]:
```
#go through every file point x seconds and get total accuracy overall
print("The overall accuracy every 0.1 secs across all wav files is {:.2f}".format(re
```

```
The overall accuracy every 0.1 secs across all wav files is 0.82
```

1.1 Prediction 2

Label each file using the majority class per all 0.1 seconds per file. Fo example is music1.wav labelled as music?

In [205]:
```
#write result for each file usually 100% correct?
if write_to_output:
    y_probs = []
    for i, row in df.iterrows():
        y_prob = fn_prob[row.file_name]
        y_probs.append(y_prob)
        for c,p in zip(classes,y_prob):
            df.at[i,c] = p
    y_pred = [ classes [ np.argmax(y)] for y in y_probs]
    df['y_pred'] = y_pred
    #get rid of the probabilities for write to file
    df = df[["file_name", "label",'y_pred']]
    #df.to_csv('predictions.csv',index=False)
df
```

Out[205]:

	file_name	label	y_pred
0	music_plus_noise1.wav	music_noise	music_noise
1	music_plus_noise2.wav	music_noise	music_noise
2	music_twintones1.wav	music_noise	music_noise
3	music_plus_speech1.wav	music_speech	music_speech
4	music1.wav	music	music
5	music2.wav	music	music
6	music3.wav	music	music
7	music4.wav	music	music
8	music5.wav	music	music
9	noise1.wav	noise	noise
10	noise2.wav	noise	noise
11	noise3.wav	noise	noise
12	noise4.wav	noise	noise
13	noise5.wav	noise	noise
14	noise6.wav	noise	noise
15	speech_nb_plus_noise1.wav	speech_nb_noise	speech_nb_noise
16	speech_nb_plus_noise2.wav	speech_nb_noise	speech_nb_noise
17	speech_nb_plus_noise3.wav	speech_nb_noise	speech_nb_noise
18	speech_nb_plus_noise4.wav	speech_nb_noise	speech_nb_noise
19	speech_nb1.wav	speech_nb	speech_nb
20	speech_nb2.wav	speech_nb	speech_nb
21	speech_nb3.wav	speech_nb	speech_nb

```
22              speech_nb4.wav       speech_nb        speech_nb
23              speech_nb5.wav       speech_nb        speech_nb
24              speech_nb6.wav       speech_nb        speech_nb
25  speech_wb_plus_noise1.wav  speech_wb_noise  speech_wb_noise
26  speech_wb_plus_noise2.wav  speech_wb_noise  speech_wb_noise
27  speech_wb_plus_noise3.wav  speech_wb_noise  speech_wb_noise
28  speech_wb_plus_noise4.wav  speech_wb_noise  speech_wb_noise
29  speech_wb_plus_noise5.wav  speech_wb_noise  speech_wb_noise
30  speech_wb_plus_noise6.wav  speech_wb_noise        speech_wb
31              speech_wb1.wav       speech_wb        speech_wb
32              speech_wb2.wav       speech_wb        speech_wb
33              speech_wb3.wav       speech_wb        speech_wb
34              speech_wb4.wav       speech_wb        speech_wb
35              speech_wb5.wav       speech_wb        speech_wb
36              speech_wb6.wav       speech_wb        speech_wb
```

All files are correctly labelled as the correct type. Doesn't agree with results below!

2 Final Predictions

Get the predictions for every x samples and compare for each file. These are the final and most appropiate results

In [197]:
```
def build_pred_per_file(fn2class,audio_dir,fn):
    y_true = []
    y_pred = []

    #for fn in tqdm(os.listdir(audio_dir)):
    rate, wav = wavfile.read(os.path.join(audio_dir,fn))
    label = fn2class[fn]
    c = classes.index(label)

    for i in range(0,wav.shape[0]-config.step,config.step):
        sample = wav[i:i+config.step]
        #sample is a block of samples x secs long
        x = mfcc(sample,rate, numcep=config.nfeat,nfilt = config.nfilt,nfft=config.n
        x = (x-float(_min))/(float(_max)-float(_min))
        x = x.reshape(1,x.shape[0],x.shape[1],1)
        #for every block of samples get the mfcc and compare to target
        #y_hat is probability for each class - softmax in final layer
        y_hat = model.predict(x)
        #y_pred is class corrsponding to max prob. for each block of samples (every
        y_pred.append(np.argmax(y_hat))
        #y_true is the real class
        y_true.append(c)

    return y_true, y_pred
```

```
In [198]: files = os.listdir(path)
          sum_scores = 0
          for file in files:
              y_true, y_pred = build_pred_per_file(fn2class,path,file)
              score = accuracy_score(y_true,y_pred)
              sum_scores += score;
              print("The overall accuracy score per 0.1 secs is {:.2f} for {}".format(score,fi

          print(sum_scores/len(files))
```

```
The overall accuracy score per 0.1 secs is 0.78 for music1.wav
The overall accuracy score per 0.1 secs is 1.00 for music2.wav
The overall accuracy score per 0.1 secs is 0.93 for music3.wav
The overall accuracy score per 0.1 secs is 0.98 for music4.wav
The overall accuracy score per 0.1 secs is 0.67 for music5.wav
The overall accuracy score per 0.1 secs is 0.92 for music_plus_noise1.wav
The overall accuracy score per 0.1 secs is 0.56 for music_plus_noise2.wav
The overall accuracy score per 0.1 secs is 0.98 for music_plus_speech1.wav
```

```
C:\Users\richard\Anaconda3\envs\tf15\lib\site-packages\scipy\io\wavfile.py:273: WavFileWarning
  WavFileWarning)
```

```
The overall accuracy score per 0.1 secs is 0.87 for music_twintones1.wav
The overall accuracy score per 0.1 secs is 1.00 for noise1.wav
The overall accuracy score per 0.1 secs is 1.00 for noise2.wav
The overall accuracy score per 0.1 secs is 1.00 for noise3.wav
The overall accuracy score per 0.1 secs is 1.00 for noise4.wav
The overall accuracy score per 0.1 secs is 1.00 for noise5.wav
The overall accuracy score per 0.1 secs is 1.00 for noise6.wav
The overall accuracy score per 0.1 secs is 0.82 for speech_nb1.wav
The overall accuracy score per 0.1 secs is 0.87 for speech_nb2.wav
The overall accuracy score per 0.1 secs is 0.96 for speech_nb3.wav
The overall accuracy score per 0.1 secs is 0.75 for speech_nb4.wav
The overall accuracy score per 0.1 secs is 0.78 for speech_nb5.wav
The overall accuracy score per 0.1 secs is 0.83 for speech_nb6.wav
The overall accuracy score per 0.1 secs is 0.98 for speech_nb_plus_noise1.wav
The overall accuracy score per 0.1 secs is 0.99 for speech_nb_plus_noise2.wav
The overall accuracy score per 0.1 secs is 0.96 for speech_nb_plus_noise3.wav
The overall accuracy score per 0.1 secs is 0.85 for speech_nb_plus_noise4.wav
The overall accuracy score per 0.1 secs is 0.87 for speech_wb1.wav
The overall accuracy score per 0.1 secs is 0.97 for speech_wb2.wav
The overall accuracy score per 0.1 secs is 0.83 for speech_wb3.wav
The overall accuracy score per 0.1 secs is 0.86 for speech_wb4.wav
The overall accuracy score per 0.1 secs is 0.95 for speech_wb5.wav
The overall accuracy score per 0.1 secs is 0.96 for speech_wb6.wav
The overall accuracy score per 0.1 secs is 0.57 for speech_wb_plus_noise1.wav
```

```
The overall accuracy score per 0.1 secs is 0.62 for speech_wb_plus_noise2.wav
The overall accuracy score per 0.1 secs is 0.62 for speech_wb_plus_noise3.wav
The overall accuracy score per 0.1 secs is 0.54 for speech_wb_plus_noise4.wav
The overall accuracy score per 0.1 secs is 0.64 for speech_wb_plus_noise5.wav
The overall accuracy score per 0.1 secs is 0.13 for speech_wb_plus_noise6.wav
0.8385432753854886
```

3 Conclusions

- We can see that the results for tend to be much better for single component audio than audio composed of two components. For example noise and speech_wb_plus_noise.
- The cause of the error is the way the files were made by me and the sample window used to collect the audio
- When the files were made I mixed two components together to generate the audio file. For example speech plus noise was constructed using speech and twin tone noise. Therefore in that file speech is continuous but varies in amplitude but the twin tone lasts only for a short time. When I take a ramdom sample of the audio file I may not be taking speech AND twin tone - It may be just speech or speech AND small component of twin tone. I believe this is the main cause of errors you see here.
- This is also seen when making predictions, two component audio results can sometimes be very good - it just depends if the sample collection is capturing the two audio components.